A method for analyzing an acoustical environment and a system to do so

The present invention departs from the needs which are encountered in hearing aid technology. Nevertheless, although especially directed to this hearing aid technology, the present invention may be applied to the art of registering acoustical signals more generically.

Current beam formers allow only weighing of incoming acoustical signals according to the spatial direction wherefrom an acoustical signal impinges on an acoustical to electrical converter arrangement.

Besides of generating such spatial angle weighing - beam forming - by means of one respectively ordered acoustical to electrical converter, it is known to provide for such weighing an array of converters, microphones, with at least two microphones. They are located mutually distant by a given distance.

For instance in the hearing aid art it is possible to adapt spatial angle dependent weighing by means of so-called beam forming, so as to eliminate noise from unwanted impinging directions. This enhances the individual's ability to perceive an acoustical signal source situated in a predetermined angular range with respect to the one or - in case of binaural hearing aid - to the two hearing aid apparatuses. Usually by such weighing function acoustical signals are primarily cancelled as impinging from behind the individual.

As current beam formers, especially on hearing aid apparatus, have only an angularly varying response, it occurs in some acoustical environments, as e.g. at a cocktail party, that even if the reception directivity is

high, the speech from a target direction is unintelligible due to superposition of different talkers located in the same direction with respect to the individual carrying the hearing aid apparatus.

It is therefore an object of the present invention to provide for a method for discriminating impinging acoustical signals not only as a function of the angular impinging direction, but also as a function of the distance of an acoustical signal's source from the hearing aid-equipped individual.

More generically, it is an object of the present invention to provide for a method and apparatus for distance-selective monitoring of acoustical signals. It is in a preferred embodiment, as especially for hearing aid apparatus, that the present invention of distance-selective registration of acoustic signals is combined with direction-selective registration of such signals.

By such combining it becomes possible to locate an acoustical source in the acoustical environment, which might be important for non-hearing aid appliances, and for hearing aid appliances it becomes possible to focus reception on a desired source of acoustical signals, as on a specific speaker.

The object of the present invention is realized by a method for analyzing an acoustical environment, which comprises

- registering acoustical signals at at least two reception locations, which are mutually distant by a given distance and generating at least two respective first electric signals representing the acoustical signal;
- calculating electronically from said first electric signals at least one of the distances of sources of

acoustical signals with respect to at least one of said locations, thereby generating a distance signal;

- amplitude filtering the distance signal, thereby generating a patterned distance signal;
- weighing a signal, which is dependent from at least one of said first electric signals by the patterned distance signal, thereby generating an output signal representing the acoustical signals from sources distributed in the acoustical environment within a distance pattern.

In a preferred mode of operation, calculation and thereby generation of the distance signal is performed according to preferred signal processing, as will be explained in more details in the detailed description part of the present description.

The second signal, which is inventively weighed by the patterned distance signal, may be directly one of the first electric signals, if only distance discrimination of an acoustical source in the acoustical surrounding is of interest. If on the other hand one desires to maintain directivity selection, then the second signal is an output signal of a directivity beam former as is known in the art and which provides for a directivity, possibly an adjustable transmission beam. Especially in view of the last mentioned combination it becomes evident that the case may arise, where selectively not only acoustical sources shall be registered in one single distance, but simultaneously from more than one predetermined distances. Therefore, the amplitude filtering may be performed with a respective filtering function, e.g. according to a comb filter, but in a preferred embodiment amplitude filtering is performed by one band-pass amplitude filtering, thereby

passing amplitude values within a predetermined amplitude band. Thereby, as the second signal is weighed, therewith only signals are output representing acoustical sources located in one distance in the acoustical environment.

As was mentioned, in a further most preferred embodiment of the inventive method, the signal dependent from the first electric signals is generated by weighing the first electric signals in dependency of the fact under which spatial angle the respective acoustical signals impinge at the at least two reception locations.

Especially with an eye on implementing the inventive method on hearing aid appliances, it is further preferred to perform amplitude filtering with an adjustable filter characteristic. Thereby and especially with an eye on providing one band-pass amplitude filtering, the individual with a hearing aid apparatus inventively construed may adjust amplitude filtering, e.g. by means of remote control, to fit to an instantaneous need of hearing, especially a specific source of acoustical signals, as a specific speaker.

In the case of the preferred implementation of the inventive method to a hearing aid apparatus or to two hearing aid apparatuses of a binaural hearing aid system, at least two microphones of the one hearing aid apparatus and/or at least two microphones, each one of the ear-specific microphones of the binaural hearing aid system, are exploited for acoustical signal reception at the at least two mutually distant reception locations.

In a further, clearly preferred realization form of the inventive method, the first electric signals are generated

as digital signals, and further preferred by additional time to frequency domain conversion.

The inventive system for analyzing an acoustical environment comprises:

- At least two acoustical to electrical converters, which are mutually distant by a predetermined distance and which generate respective first electric output signals at at least two outputs of said converters;
- a calculating unit, the inputs thereof being operationally connected to the outputs of the converters and generating at an output a signal which is representative of a distance of an acoustical source in said environment with respect to one of said acoustical to electrical converters;
- an amplitude filter unit with an input operationally connected to the output of the calculating unit and generating at an output an output signal which is dependent from a signal to the input of the filter unit, weighed by a function which is dependent from the amplitude of said input signals;
- a weighing unit with at least two inputs, one thereof being operationally connected to the output of the amplitude filter unit and the second input thereof being operationally connected to at least one of the outputs of the converters.

Further preferred embodiments or the inventive system become apparent to the skilled artisan especially by the claims 10 to 15 and the following detailed description of the invention. This especially with respect to the inventive system being implemented in a single-ear hearing aid device or in a binaural hearing aid system.

The invention will now be described more in details and by way of examples with the help of figures. They show:

Fig. 1 schematically, two reception locations mutually distant, to explain the reception characteristics enabling the inventive method and system;

fig. 2 in a simplified functional block/signal flow diagram an implementation of the inventive method at an inventive system;

fig. 3 four amplitude filter functions as preferably applied in the method or system according to fig. 2 or fig. 4;

fig. 4 a preferred realization form of the inventive method at an inventive system for directional and distance-specific discrimination of acoustical sources and as preferably implied in a single hearing aid apparatus or in a binaural hearing aid apparatus system;

fig. 5 a directivity and distance selectivity-characteristic with which S_{22} of fig. 4 depends from impinging angle and distance.

In Fig. 1 there are schematically shown two acoustical to electrical converters, microphones 1 and 2 located with a predetermined mutual distance p. If a signal source for the respective acoustical signal S_{a1} and S_{a2} is far away from the two microphones 1 and 2 and relative to their mutual distance p, there may be written:

$$S_1 = S(r_1) = S_0 \frac{1}{r_1} \exp(-jkr_1) \tag{1}$$

$$S_2 = S(r_1 + d) = S_0 \frac{1}{r_1 + d} \exp(-jk(r_1 + d)) \tag{2}$$

respectively for the electric output signals S_1 and S_2 of the microphones 1, 2. Thereby, there is valid

$$d = p\cos(\theta)\ ,\quad k = \omega/c \qquad (3)$$

p being the distance between the microphones, $\omega = 2\pi f$, with f the frequency of impinging acoustical signals S_{a1} and S_{a2}, and c the speed of sound in air.

Further, r_1 denotes the smaller one of the two distances between the respective microphones 1 and 2 and the acoustical signal source, according to fig. 1 with respect to microphone 1.

We see that the system (1) and (2) is in fact two equations of two complex values (4 equations) and the unknowns are S_0 (complex value), r_1 and d forming 4 unknowns. This means that the system is totally defined and solvable.

We then have

$$|S_1| = |S_0|\frac{1}{r_1} \qquad (4)$$

$$|S_2| = |S_0|\frac{1}{r_1 + d} \qquad (5)$$

$$\arg(S_1) = \arg(S_0) + \arg(\exp(-jkr_1)) \qquad (6)$$

$$\arg(S_2) = \arg(S_0) + \arg(\exp(-jk(r_1 + d))) \qquad (7)$$

From (4) and (5) we have

$$\frac{|S_1|}{|S_2|} = \frac{r_1 + d}{r_1} \qquad (8)$$

that leads to

$$r_1 = \frac{d}{\frac{|S_1|}{|S_2|} - 1} \qquad (9)$$

and from (6) and (7)

$$\arg(S_1) - \arg(S_2) = -\arg(\exp(-jkd)) = kd \qquad (10)$$

and then

$$d = \frac{\arg(S_1) - \arg(S_2)}{k} \tag{11}$$

and from (9)

$$r_1 = \frac{\arg(S_1) - \arg(S_2)}{k\left(\frac{|S_1|}{|S_2|} - 1\right)} \tag{12}$$

It can be observed that when the signal comes from the perpendicular of the microphone array axis, some discontinuities occur in the formulas for r_1 because in this case $|S_1| = |S_2|$ and $d=0$. If the beamforming is a 2nd order that eliminates the signal from 90°, there is no need to make a distance calculation in this direction, otherwise a third microphone can be used to perform, in the same way, the distance calculation.

In a preferred form of computation we write:

$$\frac{\rangle|S_1|\langle}{\rangle|S_2|\langle} = \left(1 + \frac{|d|}{r_1}\right) \tag{13}$$

The operator $\rangle \ldots \langle$ thereby represents an average over a predetermined time T during which the signal source may be considered as being stationary with respect to the two microphones 1 and 2. From (13) the distance r_1 becomes

$$r_1 = \frac{|d|\rangle|S_2|\langle}{\rangle|S_1|\langle - \rangle|S_2|\langle} \tag{14}$$

Therefrom, it might be seen that besides of $|d| = p|\cos(\theta)|$ r_1 may again be calculated from the two output signals of the microphones 1, 2. Nevertheless, $|d|$ too may be calculated from these output signals e.g. as will be shown. If we apply to the two signals S_1 and S_2 the function

$$G = \frac{2S_1 S_2^*}{|S_1|^2 + |S_2|^2} \qquad (15)$$

there results for kd << 1, i.e. for a distance between the microphones smaller than the wavelength of the respective acoustical signals impinging and further with d << r_1, i.e. the source being placed in a considerable distance from the two microphones

$$d \approx \frac{\mathrm{Im}[G]}{k}. \qquad (16)$$

Therefrom, there results with (15)

$$r_1 = \frac{|\mathrm{Im}[G]|}{k} \frac{\rangle|S_2|\langle}{\rangle|S_1|\langle - \rangle|S_2|\langle} \qquad (17)$$

It might be seen that r_1 is determined by the two signals S_1 and S_2 at respective frequencies f and with a predetermined distance p and may e.g. be calculated according to (17) too.

In fig. 2 there is schematically shown implementation of the findings which were explained up to now. The two output signals S_1 and S_2 of the at least two microphones 1 and 2 are input to a calculation unit 4, which e.g. according to the formulas (17) and (15) or (12) calculates the distance r_1 and generates accordingly an electric signal $S_3(r_1)$. This signal S_3 is proportional to the distance r_1. The output signal of the calculation unit 4 is applied to the input of an amplitude filter unit 6, which generates an output signal S_4 according to a predetermined filter characteristic or according to a selected or selectable dependency to the magnitude of the input signal S_3 and thus of the distance r_1.

The output signal S_4 of the amplitude filter unit 6 is applied to an input of a weighing unit 8, as e.g. to a multiplication unit, whereat at least one, e.g. the output signal S_1 of microphone 1 and as applied to a second input of the weighing unit 8, is weighed by the output signal S_4. Thereby, there is generated at the output of the weighing unit 8 a signal S_5 which accords to those parts of signal S_1 which are positively amplified by the amplitude filter characteristics of filter unit 6.

If only the components of S_1 are of predominant interest, which are generated by an acoustic signal source in one predetermined distance, the filter characteristic of amplitude filter 6 is tailored as a band-pass characteristic. Such a band-pass amplitude filter characteristic is e.g. defined by

$$F(f,r_0,r_1) = 1/\left[(r_0 - r_1)^n + 1\right] \qquad (18)$$

In Fig. 3 the attenuations F are shown for a predetermined frequency f and for $r_o = 1$, further with $n = 1, 2, 4$ and 8 respectively.

It goes without saying that the amplitude filter unit 6 is most preferably integrated in calculating unit 4 and is only drawn separately in fig. 2 for reasons of explanation.

Considering one of the amplitude filter characteristics of fig. 3 implemented as the filter characteristic of the unit 6 in fig. 2, it becomes clear that only those components of S_1 will be apparent in signal S_5, for which there is valid $r_1 = r_o$, e.g. appropriately scaled for sources with $r_1 = 1$ m.

As additionally shown if fig. 2 it is absolutely possible and often desired to have the filter characteristic of unit

6 made adjustable, so that during operation of the system one can select which area of the acoustical surrounding and with respect to distance shall be monitored.

In fig. 4 there is, still schematically, shown a preferred implementation form of the inventive method and of the inventive system, thereby especially as implied in a hearing aid apparatus or in a binaural hearing aid apparatus set. That signal processing is realized after analogue to digital conversion of S_1 and S_2 and most preferably also after time domain to frequency domain conversion, is quite obvious for the skilled artisan and is also valid at the embodiment of fig. 2. According to the specific needs, the output signal as of S_5 of fig. 2 is respectively reconverted by frequency domain to time domain conversion and subsequent digital to analogue conversion.

According to fig. 4 a matrix of at least two microphones 10 and 12 as of the two microphones of one hearing aid apparatus or of respective microphones at two hearing aid apparatuses of a binaural hearing aid system, which are distant by the respective distance p, generates the respective electric signals S_{10} and S_{12}. The electric output signals S_{10}, S_{12} are amplified, analogue to digital converted and possibly additionally filtered in units 14a and 14b. The output signal S_{14a} and S_{14b} are input to time domain to frequency domain conversion units 16a and 16b, e.g. Fast Fourier Transform units, respectively generating output signals S_{16a} and S_{16b}. In a preferred embodiment and especially for hearing aid appliances the two signals S_{16a} and S_{16b} are fed to a beam former unit 18 where, according to one of the well known calculation techniques, beam forming is realized. As schematically shown in the functional block of unit 18, the output signal S_{18}

represents principally one of the two signals S_{16}, but weighed by a function A, in fact an amplification function which is dependent from the angle θ at which the acoustical signal S_a impinges on the microphone array 10, 12.

Thus, the output signal S_{18} has a directivity selection as determined by the beam shape realized at unit 18. It must be emphasized that the present invention does not dependent from the technique and approach which is taken for realizing beam forming at the unit 18.

As was explained with the help of fig. 2, the two signals S_{16a} and S_{16b}, still representing S_1 and S_2 according to fig. 2, are input to the calculation unit 46, wherein the r_1 calculation according to unit 4 of fig. 2 and the amplitude filtering according to the function of amplitude filter unit 6 of fig. 2, are performed. The output signal of calculation unit 46 weighs at weighing unit 20 signal S_{18}. The output signal S_{22} of weighing unit 22 is frequency to time domain and digital to analogue reconverted. In a hearing aid apparatus the resulting output signal is operationally connected via the signal processing unit of the hearing aid apparatus to the electro/mechanical output converter 24 of that apparatus.

In fig. 5 there is shown the directivity and distance selection characteristic with which the signal S_{22} of fig. 4 depends from impinging angle θ as well as from distance r_1 if in unit 18 a cardioid beam former is realized, the distance between the microphones p = 12 mm and at a frequency of 1 kHz. Thereby, an amplitude filter function according to (18) was realized with r_o = 1 m and n = 2.